IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

Appl. No.: 09/732,337 Confirmation No.: 1461
Appellant: Paksoy et al
Filed: 12/07/2000
TC/AU: 2655
Examiner: Abebe

Docket: TI-28759
Cust. No.: 23494

APPELLANTS' BRIEF

Commissioner for Patents
P.O.Box 1450
Alexandria VA 22313-1450

Sir:

The attached sheets contain the Rule 41.37 items of appellants' brief. The Commissioner is hereby authorized to charge the fee for filing a brief in support of the appeal plus any other necessary fees to the deposit account of Texas Instruments Incorporated, account No. 20-0668.

Respectfully submitted,

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Rule 41.37(c)(1)(i) Real party of interest

Texas Instruments Incorporated owns the application.

Rule 41.37(c)(1)(ii) Related appeals and interferences

There are no related dispositive appeals or interferences.

Rule 41.37(c)(1)(iii) Status of claims

Claims 1-20 are pending in the application with claims 10-15 allowed, claim 3 objected to, and claims 1-2, 4-9, and 16-20 finally rejected. This appeal involves the finally rejected claims.

Rule 41.37(c)(1)(iv) Status of amendments

There is no amendment after final rejection.

Rule 41.37(c)(1)(v) Summary of claimed subject matter

Claims 1-9 relate to a wide band speech encoder with the wide band split into a higher subband plus a lower subband and the lower subband is downsampled prior to encoding; claims 16-20 relate to a corresponding decoder. The downsampling is by a factor of n/m where n and m are both integers greater than 1. Application Fig.1 and pages 2-3 describe both the encoder and decoder with an example. In the example a wide band speech signal in the 50 to 7000 Hz frequency band is split into a 0 to 5333 Hz lower subband and 5333 to 7000 Hz higher subband (filter transition 5000 to 5333 Hz). The initial sampling rate is 16 kHz, and the lower subband is extracted and downsampled by a factor of 2/3 with the steps of interpolation by 2 (increases the sampling rate to 32 kHz), lowpass filter (extract lower subband and precludes aliasing), and decimation by 3 (decreases the sampling rate to 10.67 kHz). The higher subband is extracted with a bandpass filter (5333 to 7000 Hz) and maintains the 16 kHz sampling rate. The downsampled lower subband speech is encoded with a standard narrowband speech encoder, such as a CELP coder (claim 4), and the higher subband speech is encoded with any of several possibilities (claims 5-9), such as

noise-excited LP. After transmission, the encoded lower subband speech and the higher subband speech are separately decoded and recombined to synthesize the wide band speech.

Rule 41.37(c)(1)(vi) Grounds of rejection to be reviewed on appeal

The grounds of rejection to be reviewed on appeal are:

(1) Claims 1-2, 4-9, and 16-20 were rejected as anticipated by the Nomura reference.

Rule 41.37(c)(1)(vii) Arguments

(1) Claims 1-2, 4-9, and 16-20 were rejected as anticipated by Nomura; the Examiner cited Nomura Fig.1 for the split band encoder and decoder plus Fig.2 for linear prediction (LP) coding.

With regard to claim 1, appellants reply first that the claim requires subdividing the wide band signal into a lower subband signal and a higher subband signal; whereas, Nomura extracts a lower subband signal but retains the full wide band signal instead of any higher subband signal. Explicitly, Nomura Fig.1 shows the lower subband signal at the top with downsampling and encoding by the "Bitrate Scalable MP-CELP Encoder" and shows the full wide band encoded by the "Bandwidth Extension Encoding Tool". Nomura Fig.3 details this Bandwidth Extension Encoding Tool and makes clear than the full wide band is used, not a higher subband. Consequently, Normura does not suggest the claim 1 subdividing the wide band signal into lower subband and higher subband signals.

Further, claim 1 requires the lower subband signal be downsampled by a factor of n/m where both n and m are integers greater than 1; that is, the downsampling factor is neither an integer nor the reciprocal of an integer. (The example in application Fig.1 has n = 2 and m = 3, and the downsampling changes the sampling rate of the lower subband signal from 16 KHz to 10.67 KHz.) In contrast, Nomura only suggests downsampling by a factor of 2 from the usual wide band speech sampling rate of 16 kHz to the usual narrowband

speech sampling rate of 8 KHz; see the top portion of Nomura Fig.1. This corresponds to n = 2 and m = 1, and does not suggest the downsampling of claim 1. Consequently, Nomura does not suggest claim 1 nor any of its dependent claims 2 and 4-9.

With regard to claim 16, appellants repeat the foregoing argument that Nomura does not suggest lower and higher subbands and also does not suggest the converting (upsampling) the lower subband by a factor of m/n where m and n are both integers greater than 1. Consequently, Nomura does not suggest claim 16 nor any of its dependent claims 17-20.

Rule 41.37(c)(1)(viii) Claims appendix

1. A wide band signal coder comprising:

means for subdividing signals over a bandwidth into a lower subband and a higher subband signals,

a downsampler for downsampling said lower subband signals, said downsampling by a factor of n/m where n and m are both integers greater than 1,

a low band speech coder coupled to said downsampler for encoding said downsampled lower subband signals, and

a highband coder for coding said higher subband signal without downsampling, and

a combiner for combining said higher and lower subband signals.

2. The coder of Claim 1, wherein said combiner includes a bandpass filter coupled to said highband coder to bandpass said higher subband signal to complement the lower subband.

4. The coder of Claim 1, wherein said low band speech coder is a CELP coder.

5. The coder of Claim 1, wherein said highband coder is an LPC coder.

6. The coder of Claim 1, wherein said highband coder is random noise.

7. The coder of Claim 1, wherein said highband coder is noise excited LPC.

8. The coder of Claim 1, wherein said highband coder is gain-matched analysis by synthesis.

9. The coder of Claim 1, wherein said highband coder is multi-pulse coding.

16. A wideband speech decoder system comprising:

a first decoder for decoding encoded lower subband signals;

a second highband decoder for decoding higher subband signals at a higher sampling rate than said lower subband signals;

a converter for converting said lower subband signals to the same sampling rate as the higher band signals, said converting by a factor of m/n where n and m are both integers greater than 1; and

an adder for summing said lower subband signals and said higher subband signals.

17. The decoder system of Claim 16, wherein said second decoder includes a gain-scaled random noise generator.

18. The decoder system of Claim 16, wherein said second decoder includes a gain-scaled random noise generator and the output applied to an LPC synthesis filter.

19. The decoder system of Claim 16, wherein said second decoder includes a codebook with allowable excitation vectors, a multiplier and an LPC filter.

20. The decoder system of Claim 16, wherein said second decoder includes a multipulse waveform that is gain-scaled and filtered by an LPC synthesis filter.

<u>Rule 41.37(c)(1)(ix) Evidence appendix</u>

n/a

<u>Rule 41.37(c)(1)(x) Related proceedings appendix</u>

n/a